Abstract

Throughout 1998, the TDMA community investigated and defined an enhanced suite of voice and data services for TIA/EIA-136 while also improving its basic system capabilities. These enhancements, collectively named 136+, provide improved voice quality, increased capacity, higher data rates, and improved tools for RF system engineering. The enhancements are obtained by introducing a new modulation scheme (8-PSK), new slot formats, and the addition of several new interleaving and coding options. As a result, TIA/EIA-136 now supports a new vocoder mode, the GSM Enhanced Full-Rate (US1) for improved fidelity applications, a robust IS-641 vocoder mode with a 4 dB FER enhancement on the IS-136 downlink, and a new packet data service capable of providing usable data rates of 14.4, 28.8, or 43.2 kb/s on a full-rate, double full-rate, or triple full-rate channel, respectively. The packet data service is truly evolutionary in nature, having a new medium access control layer with a network layer very similar to that used for the existing digital control channel. In addition, it maximizes commonalities among TDMA technologies, using identical higher layers and network architecture to the General Packet Radio Service specified in GSM. Furthermore, by using a concept known as tunneling to pass TIA/EIA-136 messages through the GPRS network elements, the existing features defined on the DCCH are maintained. This article provides an overview of these 136+ applications, as well as insight into additional near-term improvements, such as the ability to support six voice users per 30 kHz (TDMA6), downlink time diversity, and fast power control.

Service and System Enhancements for TDMA Digital Cellular Systems

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In the late part of 1997, the IS-136 community through the Universal Wireless Communications Consortium (UWCC) and Telecommunications Industry Association (TIA) TR45.3 adopted a three-part strategy for evolving its time-division multiple access (TDMA)-based technology toward a third-generation (3G) wireless network. The strategy consisted of enhancing the voice and data capabilities of the existing 30 kHz carrier, adding a 200 kHz carrier for high-speed data (384 kb/s) in high-mobility applications and a 1.6 MHz carrier for very-high-speed data (2 Mb/s) in low-mobility applications. The novel part of this strategy was the global convergence of IS-136 TDMA with Global System for Mobile Communications (GSM) TDMA through the evolution of the 200 kHz GSM carrier to support high-speed data applications (384 kb/s) while also improving the already efficient 30 kHz carrier for voice and mid-speed data applications. The UWCC and TIA TR45.3 successfully integrated this strategy into the TDMA 3G proposal named UWC-136 [1] submitted to the International Telecommunication Union (ITU). The focus of this article is on the enhancements (i.e., voice, data, and system named 136+) defined for the applications on the 30 kHz carrier to be published as an ANSI standard in 1999.

The 136+ enhancements to voice services are based on the commercial success of the IS-641 algebraic code excited linear predictive (ACELP) voice coder (vocoder) which was introduced into IS-136 (TIA/EIA-136) networks in 1997. The 136+ voice services program goal was to develop an even higher voice quality service, focusing on enhancements to voice quality under fading channels, high background noise, tandeming, and music conditions. The results of these efforts are the definition of two new voice services summarized in the next section.

Although a circuit-switched data service has been defined for TIA/Electronics Industry Association (EIA)-136 for some time, the TDMA community through the UWCC recognized that a packet service was also necessary. The goals of this packet service were to provide usable data rates approaching 64 kb/s in a 30 kHz RF channel, maintain existing TIA/EIA-136 coverage, and provide tight integration of the packet service with the circuit-switched service. This tight integration of the two services would allow a mobile station in operation on the packet network to automatically move to the circuit network to place and receive calls, then return to the packet network upon completion of a circuit call. It was also desirable for the mobile stations in operation on the packet network to receive key TIA/EIA-136 services such as short message service (SMS) and message waiting indications. These goals are achieved with the 136+ data service described later.

Another goal of the 136+ voice and data services is to speed the convergence of TIA/EIA-136 and GSM. This has the benefit of achieving global roaming and economies of scale for products. These goals are realizable with the joint UWCC and European Telecommunications Standards Institute (ETSI) cooperative agreement to strengthen and broaden the role of TDMA technologies for 3G systems. The agreement calls for increased exchange of information and technical documents, the appointment of observers from both the

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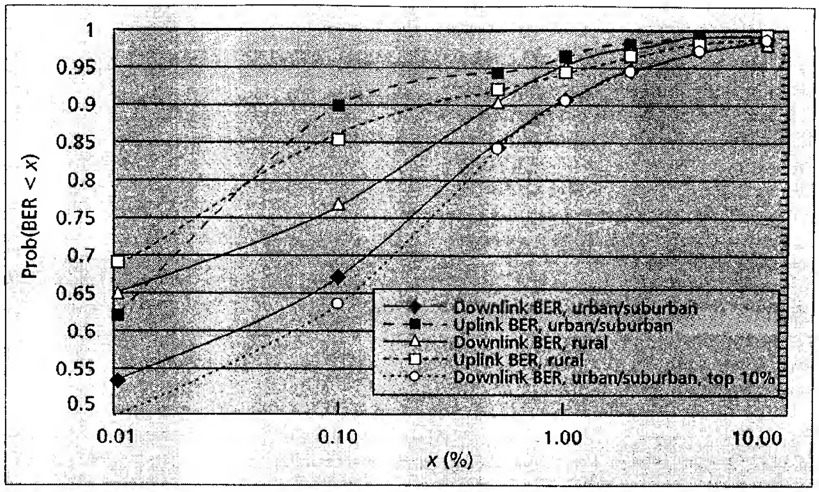
UWCC and ETSI to participate in each others' technical working groups, and the use of each organization's text, graphics, and data in the other's publications.

The aspects of the 136+ voice and data services that impact the convergence of these two TDMA technologies are also discussed herein.



■ **Figure 1**. *Uplink and downlink BER for urban and rural environments equally averaged over all cells and over the cells which carry the top 10 percent of the system load in the downlink urban environment.*

28	142	12	136	1	1	1
SYNC	Data	CDVCC	Data	F	RSVD	PRAMP

■ **Figure 2**. *CC2 downlink slot format (not drawn to scale).*

136+ Voice Services

The ACELP vocoder introduced in TDMA networks in 1997 was able to provide significant enhancements in clear channel and faded channel performance over the original VSELP TDMA vocoder. Fading channel performance improvements are of great importance since they allow a system operator to increase their quality or capacity. At the initiation of the 136+ program, it was felt that the largest opportunity for improvement in faded channels existed in the downlink for two reasons: it was perceived as the limiting link in urban areas, and uplink enhancements such as interference cancellation had recently been shown to provide large uplink gains. As an example of the first reason, Fig. 1 illustrates a bit error rate (BER) distribution recently taken from a large TDMA market in the United States. Notice that the downlink has a larger percentage of higher BERs than the uplink for the urban environment.

The 136+ voice services program focused on the development of a robust voice service mode and an improved fidelity mode. With the definition of a new slot format, improved channel encoding, and interleaving options, the robust voice service mode can achieve an additional 4 dB faded channel improvement over the existing IS-641 vocoder, as described in this section. The drivers for an improved fidelity mode include improved performance in tandeming, background noise, and music as might be encountered in wireless office applications. This improved fidelity mode is described later.

The Improved Robustness Voice Mode in 136+

The 136+ robust mode [2] is formally referred to as the Channel Coding 2 (CC2) format to differentiate it from the existing format, which is now referred to as Channel Coding 1 (CC1). One of the major differences between CC1 and CC2 is that in CC2 certain fields are removed from the downlink (base station to mobile station) slot structure to free 18 bits for use as additional channel coding. In addition, the CC2 convolutional encoder employs tailbiting and a higher-constraint-length code ($K = 7$ instead of 6) to achieve channel coding gain over CC1. Furthermore, CC2 also supports a three-slot interleaving mode for improved time diversity over the conventional two-slot interleaving employed in CC1.

CC2 Slot Structure – The 278-bit data field formed after the interleaving operation is one of several fields that together make up the CC2 slot structure. The detailed CC2 downlink slot structure is shown in Fig. 2, and consists of the following fields: a 28-bit sync field used by the receiver for synchronization purposes, a 142-bit and a 136-bit data field which together form the total 278-bit data field, a 12 bit coded digital verification color code (CDVCC) field used to minimize co-channel interference, a 1-bit fast power control (F) field used for a faster version of uplink power control, a 1-bit reserved (RSVD) field, and a 4-bit power ramp (PRAMP) field to allow time for changes in the downlink output power. The total number of bits in one slot is thus 324 bits. The major difference between the CC2 and CC1 slot structures is that the slow associated control channel (SACCH) and coded digital control channel locator (CDL) fields are not used in CC2.

The removal of the SACCH has little impact since all messages can also be sent via an FACCH message, which replaces the voice information with signaling data. Although the FACCH replaces voice, it has been shown that if the FACCH messages are either sent between talk spurts or spaced far enough in time, they are unnoticeable.

The CDL field, which previously acted as a pointer to a control channel and was formerly at the end of the downlink slot structure, is also removed. These bits were exploited since the CDL's value as a pointer is more of a last resort for mobile stations attempting to find control channels, and its loss is minimized with smarter deployment of control channels into higher probability locations where mobiles can easily find them. Deleting this field was also very important to allow downlink power control on a time slot basis from the base station. As such, it is replaced with an explicit two-symbol ramp (PRAMP) in the new slot structure.

The F and PRAMP fields were added to the CC2 format to support power control. Thus, there is a net gain of 18 data bits for the CC2 format, which are used to provide additional channel coding for the class 1a speech bits.

An Enhanced Full-Rate Vocoder – The vocoder used in the CC2 format is the same as in CC1, which is formally defined in IS-641-A [3] and often referred to as the IS-641 vocoder. The IS-641 vocoder is based on the ACELP coding model [4]. This type of vocoder is referred to as a source coder in that it attempts to replicate speech by using a priori knowledge of how speech (source) is produced. This is in contrast to waveform coders such as pulse code modulation (PCM) or adaptive dif-

ferential PCM (ADPCM), which attempt to reproduce speech by directly processing the speech waveform and subsequently transmitting the resulting signal. The specific information transmitted per 20 ms frame for the 7.4 kb/s IS-641 vocoder is shown in Table 1.

Information	Number of bits per frame
LP filter coefficients	26
Adaptive excitation	26
Fixed or algebraic excitation	68
Gains	28
Total	148

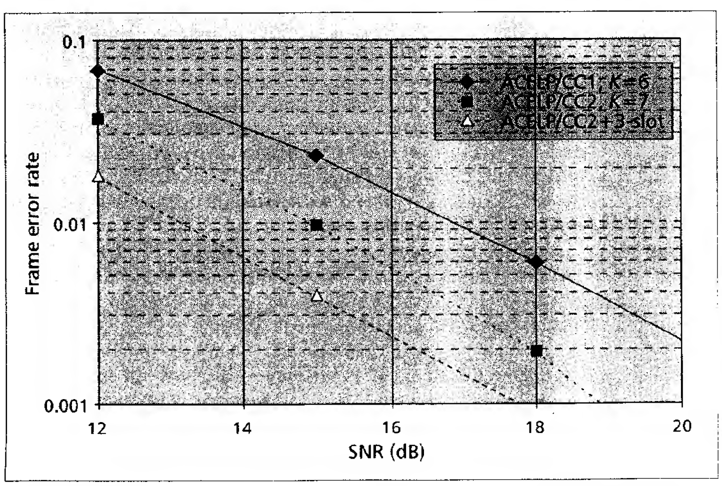
■ **Table 1.** *Bit allocation for the IS-641-A ACELP vocoder.*

Channel Coding and Interleaving – To maximize the performance of both the CC1 and CC2 formats, the vocoder output speech bits are divided into three classifications: class 1a, class 1b, and class 2. There are 48 class 1a speech bits, 48 class 1b speech bits, and 52 class 2 speech bits. The classes indicate the perceptual significance of the speech bits, with class 1a the most perceptually significant and class 2 the least. To maximize the probability of correctly receiving the most important speech bits, channel coding is applied to both the class 1a and 1b bits. Channel coding is not applied to the class 2 bits. Specifically, in CC2 a 7-bit cyclic redundancy check (CRC) is first calculated for the 48 class 1a speech bits. The resultant 55-bit stream (vector) is input to a rate 1/3 constraint length $K = 7$ convolutional encoder to produce a 165-bit output. This output is punctured by 28 bits (every sixth bit starting from the third bit is deleted) to produce a 137-bit output stream, yielding an effective code rate of approximately rate 2/5. The 48 class 1b bits are input directly to a rate 1/2 $K = 7$ convolutional encoder to produce a 96-bit output. This output is punctured by 7 bits (every 14th bit starting from the second bit is deleted) to produce an 89-bit output vector, yielding an effective code rate of approximately rate 17/32. The resultant two coded bitstreams are combined with the 52 class 2 bits to form a 278-bit input vector which is reordered and placed into a 28 x 10 block interleaver [3].

Figure 3 shows the class 1A frame error rate (FER) performance of CC2 in comparison to CC1 [5]. Notice that CC2 with $K = 7$ provides a 2 dB improvement in FER over CC1 with $K = 6$ at 10 Hz Doppler.

In addition to improving the slot format and channel coding, four interleaving formats are specified for ACELP/CC2 as shown in Table 2.

The trade-off between interleaving formats is one of delay versus link performance. The one-slot format (#1) has no interleaving delay, but requires 6 dB more link margin than the conventional two-slot format to support a 1 percent class 1a FER at 10 Hz Doppler. To ensure adequate voice quality, the one-slot format is best used for indoor applications and environments in which the carrier-to-interference ratio (C/I) and carrier-to-noise ratio (C/N) are relatively high (> 20 dB). The three-slot format (#3) is the extra robustness mode, which, in conjunction with CC2, provides about a 3.7 dB improvement in downlink performance, as shown in Fig. 3. To minimize any concerns with extra delay, three-slot interleaving is confined to only one link at a time (e.g., see formats #3, #4). This ensures that for mobile-to-mobile calls, the increase in delay over two-slot will be limited to just 20 ms. Notice that the application of format #4 adds additional time diversity with three-slot interleaving to the space diversity common in existing base stations. As such, the additional time diversity gain is about 0.5 dB less at 1 percent FER than that seen on the downlink in format #3 [6].



■ **Figure 3.** *Class 1a frame error rate performance of currently deployed systems (ACELP/CC1, K = 6), the new ACELP/CC2 downlink (K = 7), and the new ACELP/CC2 in conjunction with 3-slot interleaving.*

DQPSK Modulation – The modulation used for the CC2 format is $\pi/4$ differential quadrature phase shift keying (DQPSK) modulation, which is the same modulation used for existing TIA/EIA-136 systems. In $\pi/4$-DQPSK modulation, every 2 bits of information are encoded into one modulator symbol. The actual information is differentially encoded in the phase change from one symbol to the next. The most significant bit is the first bit in the input stream, and Gray code mapping is employed to minimize the probability of bit error.

With $\pi/4$-DQPSK modulation, the 324 input bits per slot are translated into 162 modulator output symbols. The TIA/EIA-136 symbol rate is 24.3 kHz, and thus the raw or gross instantaneous bit rate is 48.6 kb/s. In TIA/EIA-136 systems, there are six time slots per frame, and each full-rate user is assigned two out of every six time slots. Given the 24.3 kHz symbol rate and 162 symbols per time slot, each time slot is 6.67 ms long, and each frame is 40 ms long. For a full-rate user the difference between consecutive time slots is 20 ms (two out of six time slots), which is consistent with the 20 ms speech frames supplied by the vocoder. The gross bit rate for a full-rate user is one third of 48.6 kb/s, or 16.2 kb/s. This 16.2 kb/s consists of 7.4 kb/s of speech bits, 6.5 kb/s of channel coding (bit rate of the additional coding bits plus CRC), and 2.3 kb/s for the remaining fields within the time slot.

The CC1 and CC2 formats support a spectral efficiency of 1.62 b/s/Hz, based on the 48.6 kb/s gross bit rate and an effective channel bandwidth of 30 kHz.

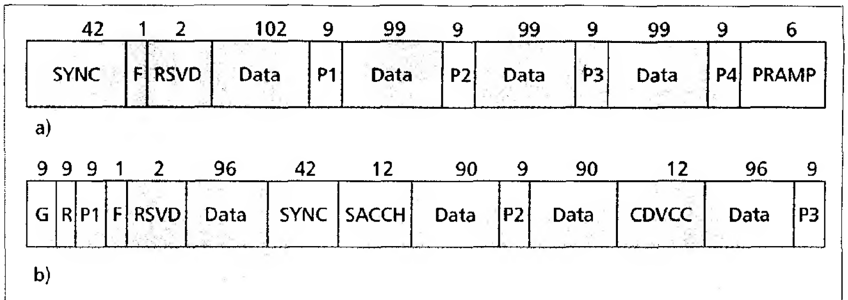
The Improved Fidelity Voice Mode in 136+

In addition to defining a more robust full-rate voice service based on DQPSK, the TIA TR-45.3 engineering subcommittee has also defined another full-rate voice service based on the US1 vocoder with 8-PSK modulation [7]. The US1 vocoder is identical to the GSM enhanced full-rate (EFR) vocoder which is also used by North American PCS-1900 operators. The US1 vocoder operates at a much higher bit rate (12.2 kb/s) than the IS-641 vocoder (7.4 kb/s), and under high C/Is and C/Ns will offer a higher quality voice service. To support this higher bit rate and still provide reasonable channel coding

within a slot requires the use of a higher M-ary form of modulation than DQPSK. A number of modulations were examined, ranging from 8, 16, 32, to 64-level constellation techniques, including differential and coherent encoding and decoding. As part of the decision, software upgradability was considered since TDMA carriers desired the ability to utilize existing radios and amplifiers with the new modulation. Consequently, most vendors believed that this could be accomplished provided the new modulation was limited to eight-level modulation and equalization requirements were relaxed to equalizing equal amplitude faded rays of 10 μs (1/4 symbol) instead of 41.2 μs as in the current IS-136 standard. These relaxed equalization requirements were believed to sufficiently cover the majority of channel conditions encountered. Furthermore, for situations with severe time dispersion, the system could always hand off the call to π/4-DQPSK modulation as a fallback option.

Once eight-level modulation was selected, a pilot-assisted 8-PSK was selected to optimize system performance at low speeds [8].

To maximize the number of bits available per slot, not only for US1 but also to prepare for the potential six users per radio frequency (RF) carrier (TDMA6), the SACCH, CDVCC, and CDL were eliminated, although their functions were maintained. For the same reasons as in the robust mode (CC2), the SACCH and CDL were deleted, whereas the explicit CDVCC in the downlink slot was replaced here by coding it into the CRC associated with the class 1a bits of the vocoder. These same fields were not deleted from the uplink slot structure, so the downlink formats are more robust by design.



■ **Figure 4.** *a) 8-PSK downlink slot format (not drawn to scale); b) 8-PSK uplink slot format (not drawn to scale).*

Format	Uplink interleaving technique	Downlink interleaving technique
#1	One-slot interleaving	One-slot interleaving
#2 (default)	Two-slot interleaving	Two-slot interleaving
#3	Two-slot interleaving	Three-slot interleaving
#4	Three-slot interleaving	Two-slot interleaving

■ **Table 2.** *Interleaving options for CC2.*

8-PSK Slot Structure – The detailed downlink and uplink slot structures for the US1 vocoder with 8-PSK modulation are as shown in Fig. 4a and 4b, respectively. In the downlink, the 399-bit data field is split into four separate data fields, three of which consist of 99 bits, with the remaining field consisting of 102 bits. The other downlink fields include a 42-bit sync field, a 1-bit F field, a 2-bit RSVD field, four 3-bit pilot fields (labeled P1–P4), and a 6-bit PRAMP field. One of the major differences between this slot structure and those of the CC1 and CC2 slot formats is the inclusion of pilot fields. These pilot fields are included to facilitate coherent detection at the receiver by developing an estimate of the channel state [9]. Since conventional 8-PSK modulation does not use differential encoding, pilot fields are required to support the channel state estimation function. For the uplink, the 372-bit data field is split into two 96-bit data fields and two 90-bit data fields. The remaining uplink fields include a 9-bit guard (G) field, a 9-bit ramp (R) field, a 1-bit -F- field, a 2-bit RSVD field, a 42-bit sync field, a 12-bit SACCH field, a 12-bit CDVCC field, and three 3-bit pilot fields. The CDVCC is also used as a pilot field since it is a known bit pattern. This has the advantage of reducing the number of dedicated pilot fields to three instead of four. The total number of bits for both the downlink and uplink is 486 bits.

In order to minimize any trunking efficiency loss due to having two modulation formats on the same carrier, it was decided that any carrier must be able to support both types of modulation. Thus, older mobiles may be assigned to a carrier, which also transmits 8-PSK channels.

Given that the previous versions of IS-136 allowed mobile stations to detect and utilize information (SYNC) from adjacent slots on the same RF carrier, the SYNC in the 8 PSK slot format remains modulated at π/4-DQPSK. Therefore, within a given time slot there are both π/4-DQPSK and 8-PSK modulations. This produces a potential problem in decoding 8-PSK since it needs a phase reference. Normally, this could be provided by simply adding a reference symbol with known phase (e.g., zero phase) placed after the differentially encoded SYNC. Unfortunately, this wastes 3 bits that could otherwise be used for data, and places the importance of the phase reference for the entire slot on the reliable detection of only one symbol.

A novel approach was used to allow channel sharing of 8-PSK time slots with π/4-DQPSK slots. The SYNC remains differentially encoded from the preceding symbol in the previous time slot, and all of its values remain the same as currently defined. After differentially encoding the SYNC from the immediately preceding symbol, a constant phase shift equal to the absolute phase of the last symbol of the immediately preceding time slot is added to all of the symbols following the SYNC. In this manner, the phase shift the receiver derives from the SYNC signal can also be removed from the data fields in order to obtain the absolute phases.

US1 Vocoder – The US1 vocoder [10] is identical in basic structure to the IS-641 vocoder. Both are ACELP vocoders, with the major difference being that the US1 vocoder uses more bits to represent the various speech parameters. The bit allocation for the US1 vocoder is summarized in Table 3. Since 244 bits are generated in every 20 ms speech frame, the resulting output bit rate is 12.2 kb/s.

Channel Coding and Interleaving – In a manner analogous to CC2, the 244 output speech bits of the US1 vocoder are divided into class 1a, class 1b, and class 2 bits. Specifically, there are 81 class 1a bits, 74 class 1b bits, and 89 class 2 bits. An 8-bit CRC is calculated based on the class 1a bits, and is appended to the original 81 bits to produce an 89-bit vector. This vector is input to a rate 1/2 $K = 6$ convolutional encoder which produces a 178-bit coded output vector. For the downlink this output is not punctured, while on the uplink six bits are punctured (every 29th bit starting at the 29th bit) to produce a 172-bit vector. Both of the resulting vectors are input to conventional 14 x 13

Information	Number of bits per frame
LP filter coefficients	38
Adaptive excitation	30
Fixed or algebraic excitation	140
Gains	36
Total	244

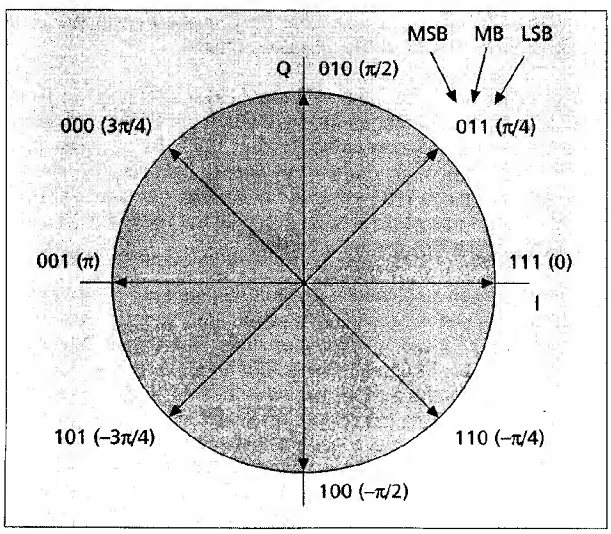
■ **Table 3.** *Bit allocation for the US1 vocoder.*

block interleavers. The extra positions that are not filled in the interleaver arrays are simply not used.

The 74 class 1b bits are input directly to another rate 1/2 $K = 6$ convolutional encoder that produces a 148-bit coded output vector. This output is punctured by 16 bits for the downlink (every ninth bit starting at the seventh bit) to produce a 132-bit vector, and by 37 bits on the uplink (every fourth bit starting at the second bit) to produce a 111-bit vector. The resulting code rates are approximately 9/16 and 2/3 for the downlink and uplink, respectively. The downlink coded class 1b bit vector is input to a 12 x 11 conventional block interleaver, while the uplink vector is input to a 12 x 10 conventional block interleaver. The downlink input fills the entire interleaver array, so all positions are used. The uplink input does not fill the entire array, and the unfilled positions are simply not used.

The coded and interleaved class 1a and 1b bits are combined with the uncoded class 2 bits to form a series of 3-bit sequences, or triads. For the downlink, there are a total of 399 bits (133 triads), while for the uplink there are a total of 372 bits (124 triads). For the downlink, the bits are combined such that every two class 1a bits (coded and interleaved) are combined with one class 2 bit to form the first 89 triads. The remaining 44 triads are formed from the class 1b bits (coded and interleaved). The bits are combined in a somewhat similar manner for the uplink. Specifically, the first 86 triads are composed of two class 1a bits and one class 2 bit. The next three triads are composed of two class 1b bits and one class 2 bit, while the remaining 35 triads are formed from just the class 1b bits. The bits are positioned within each triad to take advantage of the fact that 8-PSK modulation provides better protection to certain bit positions.

The triads are reordered to provide additional interleaving gain [7], and then intraslot interleaving is applied. As with CC2, there are three primary interleaving options: one-slot, two-slot, and three-slot. In one-slot interleaving, there is no intraslot interleaving and the current triads are simply transmitted in the current slot. In two-slot interleaving, certain triads (from the current triad vector) are transmitted in the current slot, and the remaining triads in the next slot. Thus, the current slot contains triads from the current and previous triad vector in a manner analogous to two-slot interleaving for CC2. In three-slot interleaving, the concept is extended to include another set of triads in each transmitted slot. Certain triads from the current triad vector are transmitted in the current slot, certain other triads are transmitted in the next slot, and the remaining triads are transmitted in the slot after the next slot. Thus, the current slot contains triads from the current triad vector, the previous triad vector, and the one before the previous triad vector. As with CC2, the trade-off between interleaving options is one of delay versus link performance; and as in CC2, to minimize delay in mobile-to-mobile calls the three-slot option cannot be used simultaneously on both the downlink and uplink. The final interleaved output is referred to as the *data* field and consists of 399 bits for the downlink and 372 bits for the uplink.



■ **Figure 5.** *8-PSK bit-to-symbol mapping.*

8-PSK Modulation – Conventional 8-PSK modulation is employed to transmit the information contained within each slot. A pictorial representation of the 8-PSK bit-to-symbol mapping, showing the in-phase (I) and quadrature (Q) components, is depicted in Fig. 5. As shown in the figure, each 3-bit input sequence is mapped to one of eight unique modulator output symbols. The terms MSB, MB, and LSB are defined as *most significant bit*, *middle bit*, and *least significant bit*. The output signal phase, ϕ, for every 3-bit input combination is defined in parentheses. The baseband modulator I and Q outputs are then defined simply as $\cos(\phi)$ and $\sin(\phi)$, respectively. The bits are also gray coded so that only one bit changes from one modulator symbol to an adjacent symbol. This type of coding minimizes the average BER for a given input level of C/I or C/N. It can be observed from Fig. 5 that the MSB and MB positions provide equal BER performance, which is better than that of the LSB position. This results from the fact that bit decision boundaries can be drawn for the MSB and MB which support a greater average Euclidean distance between the relevant signal points. The defined format takes advantage of this characteristic by always mapping the most perceptually significant speech bits (class 1a) into the MSB and MB positions. Likewise, the least important speech bits (class 2) are always mapped into the LSB position. As a compromise, some of the class 1b bits are mapped into the MSB and MB positions, and some into the LSB position.

A design goal of this new service was to maintain the 24.3 kHz symbol rate so that the existing transmit and receive filters (square-root raised cosine with rolloff of 0.35) could be used and the existing 30 kHz channel bandwidth maintained. The slot length of 6.67 ms and 162 symbols/slot were also to be maintained. Thus, with 3 b/symbol, 8-PSK supports 486 b/slot, which is consistent with the slot structures defined above. The instantaneous gross bit rate is 72.9 kb/s, which is a 50 percent increase over that supported by DQPSK. For a full-rate voice user employing the US1 vocoder with 8-PSK modulation, the effective gross bit rate is one third of this value (two out of six slots), or 24.3 kb/s. For the downlink, this 24.3 kb/s consists of 12.2 kb/s of speech bits, 7.35 kb/s of channel coding, and 4.75 kb/s for the remaining fields within the slot. For the uplink, the speech bit rate is the same, but the channel coding contribution is 6.0 kb/s, and the remaining fields contribute 6.1 kb/s. 8-PSK modulation supports a spectral efficiency of 2.43 b/s/Hz, again a 50 percent increase over DQPSK.

Convergence with GSM-Based Systems

Full-Rate Vocoder Similarities – As introduced previously, the speech coder selected by the UWCC and standardized by the TIA TR-45.3 engineering subcommittee to support 136+ voice services with 8-PSK modulation is the US1 vocoder. The selection of US1 for use in TIA/EIA-136 is the initial catalyst for the convergence of speech coding technologies between TIA/EIA-136 and GSM-based systems. US1 has been deployed by both PCS-1900 operators in North America and GSM 900/1800 operators in Europe, Asia, Oceania, and Africa. TIA TR-45.3 modified the original GSM 200 kHz channel coding used with Gaussian minumum shift keying (GMSK) modulation by optimizing it for the 30 kHz channel used in TIA/EIA-136.

Another potential opportunity emerging for convergence of speech coding technologies between TIA/EIA-136 and GSM systems is a result of the ETSI program associated with the development of an adaptive multirate (AMR) speech coder. The stated goal of the ETSI program was to develop a robust full- and half-rate solution to provide significant improvement in voice quality at low C/N and C/I.

The ETSI AMR program concluded the evaluation of AMR speech coders in October 1998, with the selection of an Ericsson/Nokia/Siemens candidate proposal. This AMR design selected by ETSI incorporates multiple submodes for use in full- or half-rate mode that are determined by the channel quality. The defined submodes, speech coder source rates, and channel coding rates for the ETSI AMR full-rate mode are shown in Table 4. The speech coder source rates that are common between the selected AMR design, US1/GSM EFR, and TIA/EIA-136 full-rate speech coder (IS-641) are in parentheses.

As can be seen from Table 4, the full-rate ETSI AMR has submodes that incorporate bit-exact versions of both the 12.2 kb/s US1/GSM EFR and the 7.4 kb/s IS-641 full-rate speech coders.

Six Users per Carrier: The TDMA6 Project – The UWCC initiated a TDMA6 (half-rate) program in June 1998, with defined goals, requirements, and time schedules. However, the UWCC currently does not define adaptation between full- and half-rate modes similar to the ETSI AMR, and therefore the TDMA6 program has defined coverage requirements that deviate from the ETSI AMR half-rate requirements. The TDMA6 requirements stem from two reference implementations: today's full-rate IS-641 ACELP vocoder referenced at 16 dB C/N, and a reference called TDMA6-641 which denotes an implementation of 6 users/carrier using the IS-641 ACELP vocoder and 8-PSK. The minimum performance criterion for TDMA6 candidates is that they must be better than the best of either of the two references for a given C/N. The TDMA6-641 reference has been shown to have essentially the same performance as today's full-rate (i.e., ACELP/CC1) albeit with a 5 dB penalty. Therefore, above 16 dB C/N a TDMA6 candidate must be better (e.g., using mean opinion score, MOS, scoring) than today's full-rate at 16 dB, and it must be statistically equivalent or better than TDMA6-641. The difference in these requirements from other traditional half-rate programs is that the performance at high C/Ns (e.g., > 26 dB) must converge to the same performance as today's full-rate ones (i.e., landline quality).

Based on the defined requirements, it is evident that the vocoder source rate should be maintained at an optimal level with appropriate channel coding applied to minimize the degradations encountered in the mobile radio environment. Thus, a TDMA6 design involves a trade-off between vocoder bit rate and the amount of channel coding required to combat the degradation encountered in the radio channel. The TDMA6 program is scheduled to conclude in mid-1999 with the selection of a candidate proposal. It is envisioned that at least one proponent will submit an ETSI AMR-based solution. The defined submodes, speech coder source rates, and channel coding rates for the ETSI AMR half-rate speech coder are shown in Table 5.

Submode	Speech coder source rate	Channel coding rate
1	12.2 kb/s (US1/GSM EFR)	10.6 kb/s
2	10.2 kb/s	12.6 kb/s
3	7.95 kb/s	14.85 kb/s
4	7.4 kb/s (IS-641)	15.4 kb/s
5	6.7 kb/s	16.1 kb/s
6	5.9 kb/s	16.9 kb/s
7	5.15 kb/s	17.65 kb/s
8	4.75 kb/s	18.05 kb/s

■ **Table 4.** *ETSI AMR full-rate submodes.*

As one can see, the ETSI AMR half-rate vocoder currently includes a bit-exact version of the IS-641 7.4 kb/s full-rate speech coder as one of its five defined submodes.

Enhanced Reporting Capabilities and System Improvements

MAHO – Several of the reporting capabilities were also enhanced in the next revision of IS-136. The mobile-assisted handoff (MAHO) function requires the mobile to perform signal strength measurements on up to 24 neighboring cells and continuously transmit a low-pass filtered version to the system, typically using the SACCH channel. In addition, the report includes estimated BER on the assigned channel. The introduction of hierarchical cell structures and, in particular, indoor use of cellular have resulted in the demand for higher dynamic range of the reported signal strength measurement. Consequently, strong received signals are now truncated at a 20 dB higher level than previously.

Furthermore, the BER resolution has been improved and two completely new functions were introduced as part enhanced MAHO reporting. The mobile can now be instructed by the system to also include either a C/I or a word error rate (WER) estimate of the assigned channel. Note that WER and FER are actually the same quantities. A reference C/I estimation algorithm [11], using the known values in the SYNC and CDVCC fields (when present), is provided in the specification. The reference algorithm is quite accurate, able to provide C/I estimates within 1 dB in less than 1 s, and any algorithm used must be as accurate or better than the reference. The WER estimate reports values between 0–64 percent WER and is based on the CRC decoding associated with the voice coder or FACCH signaling.

MACA – The mobile-assisted channel allocation (MACA) function requires the mobile to perform signal strength measurement on up to 16 RF carriers, and BER, WER, and signal strength on the current control channel while in sleep mode. The measurements are averaged and reported to the system when making an access (e.g., originating or terminating call event). These measurements can be used by the system for the subsequent traffic channel allocation or as a radio network performance tool monitoring the perceived quality of the downlink control channel and general cell-planning layout. The dynamic range of the MACA report was similarly extended.

Downlink Power Control – In addition to the above reporting enhancements, the next revision of IS-136 also allows for

Submode	Speech coder source rate	Channel coding rate
1	7.95 kb/s	3.45 kb/s
2	7.4 kb/s (IS-641)	4.0 kb/s
3	6.7 kb/s	4.7 kb/s
4	5.9 kb/s	5.5 kb/s
5	5.15 kb/s	6.25 kb/s
6	4.75 kb/s	6.65 kb/s

■ **Table 5.** *ETSI AMR half-rate submodes.*

downlink power control on a time slot basis. Current deployed systems have constant power on the downlink RF carrier provided that any one time slot is active. This has some advantages for the mobile stations since they can utilize adjacent time slots to aid in synchronization or equalization. However, this comes at the expense of extra interference, and the loss of per-slot power control capability. There are now two types of per-slot power control accounted for:

- Single antenna per-slot power control as would be generated by a base station using one transmit antenna
- Smart antenna per-slot power variation as would be generated from a base station using multiple smart antennas

New signaling is defined to inform the mobile of which type of per-slot power control is being used. This signaling is provided since the mobile station may still desire to use information from adjacent time slots when per-slot power control is not activated or single antenna per-slot power control is utilized. The mobile station may not want to use information when smart antennas are used since signals in adjacent slots may not be correlated, given they can be sent from different antennas.

Downlink power control (DPC) on a time slot basis is allowed on all radio channels except those that contain control information since they are used as beacons by the mobile station for finding the best cell site. The base station is allowed to ramp up/down in the last two symbols in the downlink time slots, as shown in Figs. 2 and 4. Since there may be older mobile stations on the same RF carrier as the new mobile station, the base station is recommended to perform per-slot power control provided only new mobile stations are on a given RF carrier. New mobile stations are notified as to the change of downlink power state of a given RF carrier changes. For example, should a mobile not knowledgeable of per-slot power control be placed on a channel that was previously power controlled, all existing mobile stations on this RF carrier will be notified that the RF carrier has been switched to no power control. In order to maintain fast channel acquisition, the standard dictates that upon handoff or initial channel designation, the base station should ensure that the power of the assigned time slot is no lower in power than any other time slots on that RF carrier. To ensure that per-slot power control does not significantly degrade the performance of automatic frequency planning algorithms which utilize channel measurements from existing mobiles, a new form of MACA channel measurement using a "peak hold" technique was also incorporated. The peak hold concept makes sure that the mobile station measures across all the slots on a given RF channel and reports back the strongest time slot.

Current Standardization Efforts

Downlink Time Diversity with ADVICE – Another topic currently under standardization in TIA TR45.3.5 is the Added Diversity for Improvements in Channel Errors (ADVICE) project. This also is a method to improve the downlink quality by activating additional time slots (ADVICE slots) when the link quality suffers toward a given user. In this manner, the mobile station receiving the ADVICE slots can reduce the radio link error rate and thus improve voice quality. Provided these slots are relinquished when newly arriving calls on the cell can find no other free time slots, no capacity loss is encountered.

To assess the performance gain, several methods were reviewed. Although it is possible to send additional vocoder bits in the adjacent time slots, all of the methods studied either consisted of treating the ADVICE slots as punctured bits from the encoding used in the assigned slots [12, 13], or setting the ADVICE slots as novel "repeats" of the assigned slots [14]. Although the former methods provided the best performance, they were slightly more computationally intensive, and would also require forward error correction definitions for each type of service (vocoder, data) that is desired to benefit from ADVICE. Consequently, a repetition method was chosen, which utilizes some of the concepts from space-time coded modulation (STCM) [15]. In this technique, the STCM concept of transmitting symbols sequentially over two different antennas is replaced here instead by mapping the symbols over two different time slots. Thus, this can be thought of as time-time coded modulation (TTCM). The same mapping sequence as described in the STCM work [15] is used. For example, if value 0 is transmitted in slot 1, value 0 is transmitted in the same symbol position in slot 2. If value 1 is transmitted in slot 1, value 5 is transmitted in the same position in slot 2, and so on, as shown Table 6.

Figure 6 shows the simulated diversity gain at 1 percent BER of TTCM assuming all of the bits in time slots 2 and 5 can be combined via maximal ratio combining with those of time slots 1 and 4 [14].

The slot normalized fading rate in Fig. 6 is equal to the Doppler frequency multiplied by the time for one slot (i.e., 6.67 ms). For comparison, Fig. 6 also depicts the diversity gain due to maximal ratio combining of the signal from a straight slot repeat. In comparison, it is seen that the TTCM scheme outperforms simple repeat by about 1 dB at low speeds and close to 2 dB at high speeds. Additionally, it is noted that if pre-slot antenna diversity is utilized a diversity gain of over 7 dB can be obtained across all speeds.

These link-level gains suggest that when ADVICE is turned on, significant improvements can be realized. Intuitively, idle time slots nearly always exist due to the fact that most operators run their networks at low blocking percentages (e.g., 2–5 percent). Additionally, since most operators design their systems such that the signal quality is less than acceptable only a small percentage of the time, the need for ADVICE should only occur for a small percentage of the users. For example, in a theoretical $N = 7/3$ sector reuse pattern, only 10 percent of the time is the signal quality less than acceptable (C/I < 17 dB). Simulations were performed to investigate the percentage of time ADVICE slots, using only idle slots, are available for use [16]. For simplicity, the simulations assumed that 10 percent of the calls needed ADVICE, that

8-PSK value		Value in time slot 1, 2
0	→	0, 0
1	→	1, 5
2	→	2, 2
3	→	3, 7
4	→	4, 4
5	→	5, 1
6	→	6, 6
7	→	7, 3

■ **Table 6.** *8-PSK symbol mapping for the ADVICE concept.*

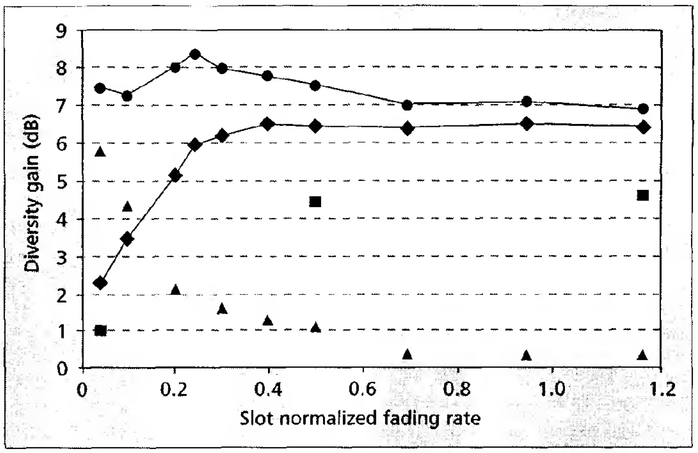
ADVICE was needed for the duration of the call, and that call arrivals could be modeled as Poisson with average call duration of 120 s. Under 15 and 5 RF channels, ADVICE was shown available 62 and 73 percent of the time when needed, respectively, and could be maintained 63 and 85 s, respectively, before it needed to be released to allow new calls onto the system (i.e., not increase blocking).

There are applications for using ADVICE in both coverage- and interference-limited environments. In coverage-limited environments, although link balance is generally strived for and the norm, uplink performance is sometimes better than downlink performance. This happens in cases such as when tower-top amplifiers are added to an existing cell site, an RF engineer has to sacrifice a design using smaller transmit coax cables in order to collocate on an already crowded tower, a particular site has better diversity gain than normal, lower output power amplifiers are used, or possibly more diversity branches (e.g., four-branch diversity) are added to the cell site.[1] In all of these cases, unintentional downlink coverage holes surface, which with the use of ADVICE could be filled in provided the performance gains are sufficient.

Understanding the benefit of ADVICE in an interference-limited environment is somewhat more difficult since one needs to consider it in conjunction with other link-quality improvement techniques such as handoff based on poor link quality (BER). Handoff based on poor link quality (BER, C/I, FER) is used today in hopes that another RF channel will have lower interference than that present on the current channel. The inherent problem with handoff based on poor link quality, however, is that when the interfering cells are at near full load, it doesn't matter to which channel a handoff is made; it will most likely be interfered with as well.

Figure 7 shows the cdf of the C/I in the desired cell under three different link quality managing techniques (i.e., handoff, ADVICE, handoff+ADVICE) with five RF channels serving 4.5 Erlangs of traffic (i.e., half load of 2 percent blocking) assuming that frequencies from interfering cell sites are active with a probability of 85 percent. ADVICE gain is assumed to have a 3 dB improvement in C/I, and is activated only after handoff based on BER failed to improve the link quality (i.e., C/I < 15 dB). Notice that the full 3 dB (at 1 percent C/I point) is obtained over simple handoff based on poor link quality.

This illustrates that when the load on the interfering cell sites is high, ADVICE improves the link quality. As the load increases on the interfering cells, the value of link quality based handoff decreases toward zero. ADVICE provides an additional tool to keep the performance gains independent of load.



■ **Figure 6.** *Time diversity gain at 1 percent BER. Diamond: TTCM; square: simple repetition code; triangle: selection diversity between two slots; circle: combined TTCM and preslot selection diversity.*

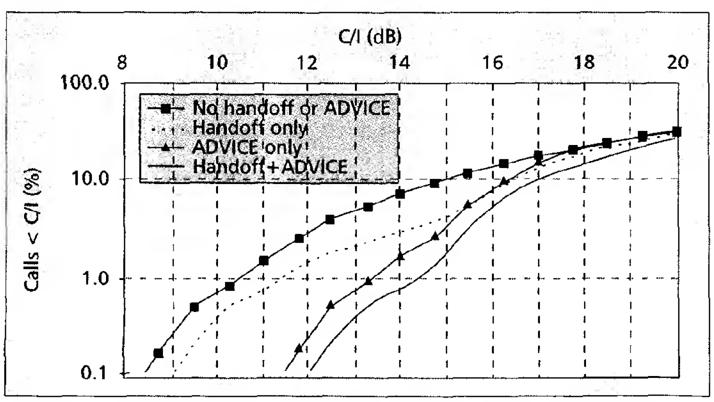
Fast Power Control – All of the new slot formats contain at least one reserved bit for fast power control. TIA TR45.3.5 is currently investigating algorithms to utilize this bit to improve the average C/I performance over a cell. Given that the bits occur once every 20 ms, power commands are possible 50 times a second. In comparison with the power control available today which is once every few seconds, this is quite a bit faster; however, it is still not fast enough to track the Rayleigh fading, and improvements are thus confined to reducing the shadow variations. Nevertheless, contributions [17, 18] have shown that the performance can be significantly improved (3–4 dB in average C/I) despite even a 10 percent error rate in the power control bit. Standards setting for this feature is expected during 1999.

136+ Packet Data Service

An Overview of GPRS-136

The UWCC and TIA TR-45.3 decision to use technology based on Enhanced Data for GPRS Evolution (EDGE) as the UWC-136 solution for the high-speed data component of IMT-2000 prompted the selection of the General Packet Radio Service (GPRS) network architecture for supporting 136+ packet data and 3G packet transport. Perhaps the most important factor influencing the selection of the GPRS network architecture was the compatibility and synergy with EDGE network capabilities. Use of the GPRS network architecture for 136+ packet data service enables data subscription roaming with GSM networks around the globe that support GPRS and its evolution. The 136+ packet data service standard is officially known as GPRS-136.

GPRS-136 provides the same capabilities as GSM GPRS. The user can access two forms of data network, X.25 and Internet Protocol (IP)-based. The user may have quality of service (QoS) requirements for any data session, and for IP-based networks the user may have dynamic or



■ **Figure 7.** *C/I distribution with respect to the three link quality management techniques (handoff, ADVICE, handoff+ADVICE).*

[1] *Four-branch diversity is now available from several infrastructure vendors.*

Modulation and link	Full-rate	Double-rate	Triple-rate
π/4-DQPSK uplink	13.10 kb/s	26.20 kb/s	39.30 kb/s
π/4-DQPSK downlink	12.80 kb/s	25.60 kb/s	38.40 kb/s
8-PSK uplink	17.85 kb/s	35.70 kb/s	53.55 kb/s
8-PSK downlink	17.40 kb/s	34.80 kb/s	52.20 kb/s

Table 7. *GPRS-136 peak data rates at the physical layer.*

static IP allocation. The user may have multiple data sessions in operation at one time. These sessions are called *Packet Data Protocol* (PDP) *contexts*. The number of PDP contexts that are open for a user is only limited by the user's subscription and any operational constraints of the network. The main goal of the GPRS-136 architecture is to integrate TIA/EIA-136 and GSM GPRS as much as possible with minimum changes to both technologies. In order to provide subscription roaming between GPRS-136 and GSM GPRS networks, a separate functional GSM GPRS home location register (HLR) is incorporated into the architecture in addition to the TIA/EIA-41 HLR. The general approach of the GPRS-136 data model is to overlay the circuit-switched network nodes with packet data network nodes for service provisioning, registration, mobility management, and accounting. However, interworking is provided between the circuit-switched and packet data networks for mobiles capable of both services (Fig. 9). Furthermore, unlike CPDP, GPRS-136 allows a user engaged in an active data transfer to suspend operation should he/she wish to make or receive a call. Depending on the data operation being performed, the data transaction may be resumed once the voice call is completed.

GPRS-136 shares a common physical layer with TIA/EIA-136, and is capable of supporting full-, double-, and triple-rate TDMA operation on a 30 kHz RF channel. Two modulation types are supported: π/4-DQPSK and 8-PSK. Table 7 summarizes the GPRS-136 peak data rates available for the different operational modes.

Network Architecture

Figure 8 shows the GPRS-136 functional architecture. The core packet part of the network consists of the serving GPRS support node (SGSN), gateway GPRS support node (GGSN), equipment identity register (EIR), GPRS HLR, and packet data network (PDN). These network elements are defined the same as in GPRS, with minor modifications to the SGSN to support TIA/EIA-136. The GPRS HLR is a GSM HLR that supports GSM authentication and subscriber data management, as well as SGSN mobility management. The other functional elements in the network architecture are defined in the TIA/EIA-41 standard. The gateway mobile switching center (MSC) reduces the impact on the serving TIA/EIA-41 MSC by supporting the new Gs′ interface. However, the gateway and serving MSCs may be integrated in a commercial deployment.

The following subsections give a more detailed explanation of the functions in Fig. 8. For complete explanations, the reader is referred to GSM documents [19].

Functional Elements – The SGSN is the node serving the mobile station (MS) and is responsible for routing data packets to the correct GGSN. The SGSN is responsible for communicating with the HLR and optionally an access point name (APN) server to determine which GGSN to use. The node then establishes a PDP context with a GGSN. The PDP contains information identifying the external data network type (X.25 or IP) and, in the case of IP, if a dynamic or static address is to be used. The PDP context may also contain information stating the QoS required by the MS. The node is also responsible for performing authentication of access attempts and encryption of user data over the air interface.

The GGSN is the node that is accessed by the external PDN to deliver data to the user. It contains routing information for attached GPRS users. The routing information is tunneled to the MS's current point of attachment (i.e., the SGSN). The GGSN may also be the node that provides a dynamic IP address for a requesting MS.

The GPRS HLR is the main subscriber database for GPRS-136 data services. Packet data service information specific to a subscriber and his/her MS is held in the GPRS HLR in GSM format. This includes the subscriber's current location, temporary network address, dynamic and/or static IP addressing capabilities, and the services he/she is allowed to access. GSM authentication information is also stored in the GPRS HLR. The location of the GPRS HLR depends on subscribers' numbers and may be standalone, or collocated with the TIA/EIA-41 HLR or with a GSM HLR.

The APN is the node that may be accessed by the SGSN to determine which GGSN the MS wants to use. It is equivalent to an Internet Domain Name Server.

The gateway MSC is a logical entity that interfaces the GSM GPRS core network to the TIA/EIA-136 and TIA/EIA-41 network so that integrated voice and data services can be provided. The functionality may be incorporated into either the TIA/EIA-41 MSC, the SGSN, or a standalone element.

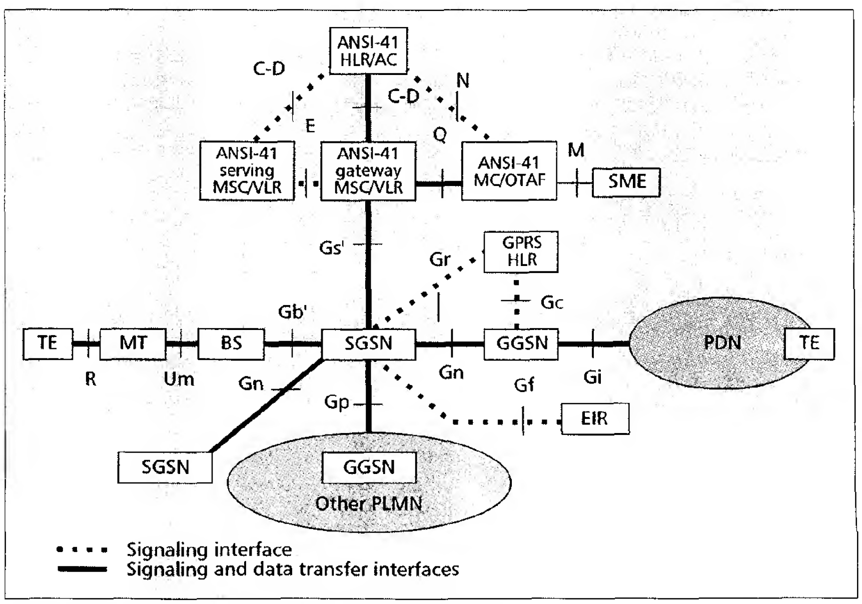
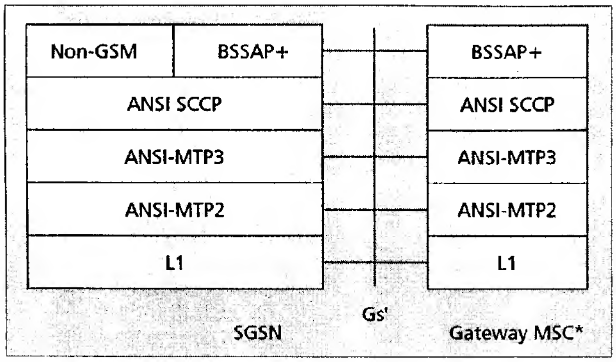


Figure 8. *GPRS-136 functional network architecture.*



■ **Figure 9.** *The Gs′ protocol stack.*

As in GSM, different GPRS-136 MS classes are defined to address different applications in the marketplace. Two GPRS-136 MS classes are defined: class B136 and class C136. A class B136 MS has characteristics similar to those of a class B GSM MS. It can invoke packet calls or circuit calls sequentially, but not simultaneously. A class C136 MS supports GPRS-136 packet services and limited TIA/EIA-136 functions that do not include circuit calls.

Network Interfaces – In addition to functional elements, Fig. 8 shows a number of open GPRS-136 interfaces. The majority of these interfaces are as defined in the relevant GSM, GPRS, and TIA/EIA-41 standards. The Gs′ interface and access network interfaces are unique to GPRS-136. The access network includes the interface from the SGSN to the base station (BS), called the Gb′ interface, and the BS to the MS, called the Um interface.

Figure 9 shows the Gs′ interface stack used with GPRS-136. The GSM Gs interface [19] is an optional enhancement to the GSM MSC that enables efficient coordination of GPRS and non-GPRS services and functionality. Typical functionality supported by the Gs interface includes non-GPRS paging, non-GPRS location updates, non-GPRS alerts, and International Mobile Station Identity (IMSI) detaches. GPRS-136 uses a subset of the Gs interface, along with a tunneling capability to transfer TIA/EIA-136 layer 3 mobility management and teleservice messages transparently between the TIA/EIA-41 MSC and MS through the SGSN.

Figures 10 and 11 show the protocol stack for the Gb′ interface. Figure 10 shows the same protocol stack found in the GSM stage 2 recommendation [19]. Figure 11 is an extension to this protocol stack that includes another layer on top of the GSM logical link control (LLC) to transport the TIA/EIA-136 layer 3 messages between the gateway MSC and the MS transparently through the SGSN. This layer is known as TIA/EIA-136 mobility management, or 136MM. On the SGSN side, this has required two new service access points (SAPs) to be defined to carry the 136MM messages, one SAP to support high-priority information, such as mobility management, and another SAP for lower-priority messages such as teleservices.

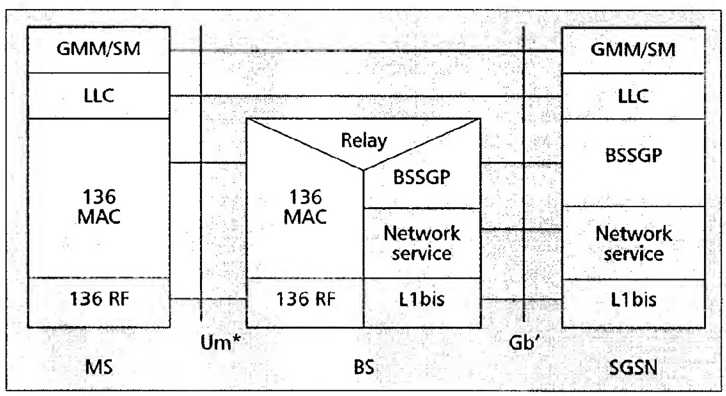
The Air Interface

The air interface design of GPRS-136 allows a GPRS-136 MS operating on a 30 kHz channel to access and obtain service from a GPRS network. Figure 12 shows an expanded view of the lower layers of the protocol stack for GPRS-136. The subsections that follow describe several air interface entities shown in Fig. 12.

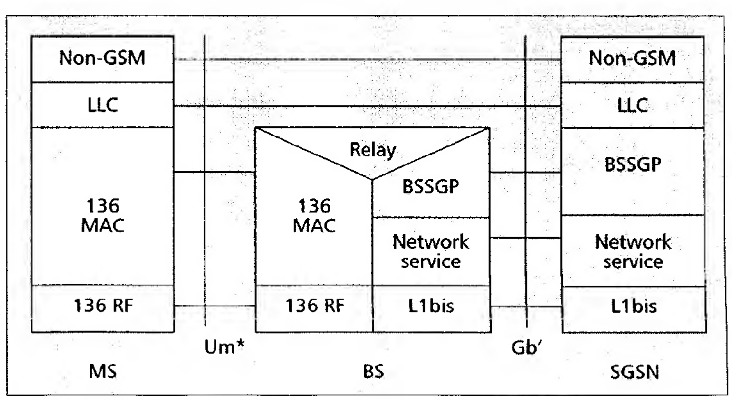
Physical Layer – The physical layer of GPRS-136 has been designed to be highly compatible with the physical layer for the voice services of 136+. In GPRS-136, a packet data channel (PDCH) can be configured to be full-rate and can share the same RF channel with a digital control channel (DCCH), digital traffic channel (DTC), or another PDCH. A PDCH can also be configured to be double-rate and triple-rate in order to provide higher data rates to the MS user. The physical layer slot structure for GPRS-136 has been designed to be very close to the slot structure for 136+ voice services.

The superframe and hyperframe structures for GPRS-136 are also identical to those of the TIA/EIA-136 DCCH (Fig. 13). As in the DCCH random access control channel (RACH), subchanneling exists on the uplink random access channel on the PDCH. Each full-rate PDCH is defined to consist of three subchannels, as opposed to six subchannels on a full-rate DCCH. On a full-rate PDCH, the three subchannels create three distinct access paths for mobiles to access the PDCH. Paging frame class (PFC) and sleep mode are defined to be similar to the DCCH.

Medium Access Control – The MAC sublayer manages access to physical-layer resources to minimize collisions between multiple users and to efficiently use the RF resources. The MAC sublayer serves as a shared medium between multiple MSs and the BS for the transfer of higher-layer service data units (SDUs). Higher layers that utilize the services of the MAC sublayer are LLC, RRME, and BME. The following modes of operation are used to support delivery of data for these higher layers:



■ **Figure 10.** *The MS-to-SGSN GSM signaling plane.*



■ **Figure 11.** *The MS-to-SGSN non-GSM signaling plane.*

- Unacknowledged mode — Supported only on the downlink for LLC and RRME requests
- Acknowledged mode —Supported on the uplink and downlink for LLC and RRME requests
- Broadcast mode — Supported on the downlink for BME requests

In acknowledged mode, the MAC is responsible for in-sequence delivery of data to the higher layers. Error recovery is handled using a sliding window automatic repeat request (ARQ) protocol. Most transactions are expected to be in acknowledged mode with the exception of short transactions, such as delivery of pages.

The physical layer resource used by the MAC sublayer may consist of one or more PDCHs. Each PDCH consists of some or all of the following logical subchannels:

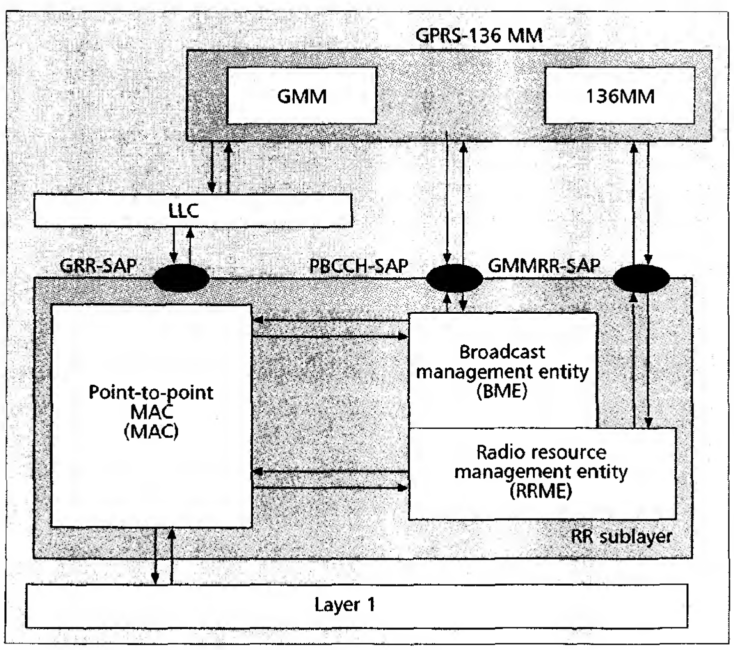
- PRACH — Packet random access channel, the uplink logical subchannel used for all mobile accesses and uplink data delivery.
- PBCCH — Packet broadcast channel, the downlink logical subchannel used to carry system broadcast information and broadcast SMS information. The PBCCH further comprises the fast-PBCCH (F-PBCCH), extended-PBCCH (E-PBCCH), and SMService-PBCCH (S-PBCCH).
- PPDCH — Packet paging and data channel, the downlink logical subchannel used to deliver pages and data payloads.
- PCF — Packet channel feedback, the downlink logical subchannel used to provide feedback for contention- and reservation-based access on the PRACH.

GPRS-136 defines two types of PDCHs:

- Packet control channel (PCCH) — This type of PDCH supports all the above-listed logical subchannels. It supports contention- and reservation-based accesses on the uplink.
- Packet traffic channel (PTCH) — This type of PDCH supports only PRACH, PPDCH, and PCF subchannels. PBCCH, contention-based access, and mobile sleep mode are not supported.

The GPRS-136 MAC sublayer supports transactions with different priorities. The MAC sublayer provides two SAPs, called MAC logical-link identifiers (MLLIs), to the higher layers, where MLLI = 0 has a higher priority than MLLI = 1. Whenever a higher-layer entity requests the service of the MAC sublayer, it provides the MAC with the payload SDUs, QoS information, and Priority parameter. The QoS information indicates which MLLI the MAC should use. Time-critical transactions, such as control messages, always use MLLI = 0, whereas regular data transactions may use MLLI = 1. In addition to the two MLLI priority levels, the Priority parameter allows four levels of priority. For uplink transactions, the MS may request a priority for its transaction by setting this Priority parameter to the appropriate level in the Mobile Priority Class field of the BEGIN frame. For downlink transactions, the BS MAC sublayer considers this Priority parameter when it schedules slots for a downlink transaction.

GPRS-136 has some unique design features such as active mobile identity (AMI) management, implicit addressing, adaptive modulation, and adaptive channel coding (incremental redundancy mode), which have not been described here in detail due to dearth of space.



■ **Figure 12.** *Air interface entities of GPRS-136.*

Radio Resource Management – Radio resources in the GPRS-136 system are managed by two logical entities: the RRME and BME. The BME is responsible for the transmission of broadcast messages at the BS and the reception of these messages at the MS. The RRME is responsible for PCCH selection/reselection, radio link quality monitoring, and channel assignment procedures.

The goals of the GPRS-136 radio resource management procedures are:

- To allow a GPRS-136 mobile to acquire usable GPRS-136 service within an acceptable period of time even if it has no prior knowledge of PCCHs in the system
- To provide broadcast information to MSs
- To spread the load of MSs across multiple PCCHs if they are available for allocation, and to distribute the paging load within a given PCCH by spreading all camped mobiles evenly across all available paging time slots
- To be able to provide continuity of service to a user even when the user is mobile by enabling the MS to reselect the best available PCCH
- To reassign active MSs to PCCHs or PTCHs other than the PCCH originally selected for camping

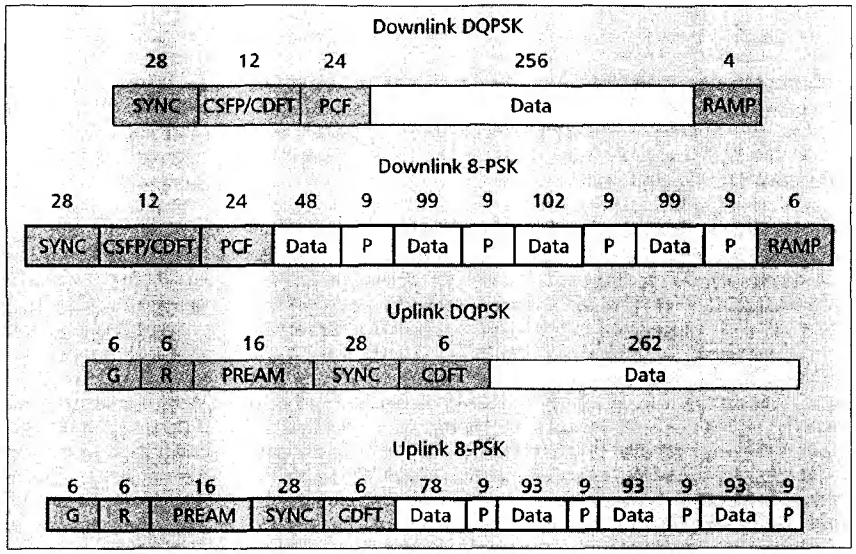
On power-up, a GPRS-136 MS reads the broadcast information on the available DCCH, which indicates a beacon PCCH if available. A GPRS-136 MS is mandated never to camp on a PCCH without going through a DCCH. Once the MS has read the beacon PCCH Parameters message on the DCCH, it invokes PCCH selection procedures for camping purposes.

In order to maintain similarity with the DCCH, many GPRS-136 radio resource management procedures have been designed to be the same as the DCCH. The PCCH channel assignments are exactly the same as for the DCCH. The RRME in the BS controls the reassignment of PDCH resources and may do so at any time during a transaction. The Signal Strength Aspects Determination procedure for GPRS-136 has been derived from the DCCH Signal Strength Aspects Determination procedure. The accuracy of signal strength measurements is also based on the accuracy specified for the DCCH.

Once a GPRS-136 MS has entered the camping state on a PCCH, it can reselect another

PCCH. This reselection can take place if there is a radio link failure, if the cell is barred, if there is server degradation, if there exists a priority system condition, and as the result of a periodic evaluation of neighboring channels. Reselection may also occur while an MS is active on a PTCH.

Monitoring of radio link quality (MRLQ) is performed at the MS in both sleep and active modes. In sleep mode, the MS estimates radio link quality by measuring the coded superframe phase (CSFP) decoding errors based on the superframe phase value it expects when it wakes up in its paging slot. In active mode, the MS also estimates radio link quality by measuring the CSFP decoding errors, but in this case the CSFP field transmitted in consecutive slots is decoded for the estimation.



■ **Figure 13.** *Different modulation schemes and their frames.*

Mobility Management – The mobility management functions of GPRS-136 ensure that the network knows the current location of MSs and provides user identity confidentiality. Since the GPRS-136 system is an integration of the GSM-based GPRS system and the TIA/EIA-136 system, it has two mobility management entities operating in parallel. These are the GPRS mobility management (GMM) and 136MM entities.

The GMM entity performs procedures for packet data mobility management for an MS obtaining service from the GPRS-136 packet system. The 136MM entity performs mobility management procedures for an MS obtaining non-GPRS circuit-switched services from the TIA/EIA-41 side of the network. The interaction between an MS and the TIA/EIA-41 network elements is done by tunneling the messages transparently through the SGSN to the TIA/EIA-41 MSC. This tunneling concept is a key element of GPRS-136 mobility management.

GPRS-136 GMM procedures are exactly those defined in GSM 04.08, except that both class B136 and class C136 MSs are mandated to follow only the GMM procedures applicable to GPRS class C MSs. This is because for Class B136 MSs, mobility management messages related to circuit-switched services are tunneled through the SGSN. 136MM procedures for GPRS-136 pertain to non-GPRS circuit-switched services and are applicable to both class B136 and class C136 MSs.

GMM and 136MM are closely related entities. A typical interaction between them is a request from 136MM to GMM to initiate a GMM procedure such as Attach, Routing Area Update, Detach, or Suspend. The GMM response is to notify 136MM with a success or failure indication, triggering a 136MM procedure or action. When an MS is on the packet system, the GMM entity is the primary mobility manager with respect to the packet system and 136MM is a client of the LLC layer that uses its services through the tunneled TIA/EIA-136 signaling messages.

Summary of Operation

In the discussion to follow, a combined gateway/serving MSC is assumed. On power-up, a GPRS-136 MS first scans for a DCCH. If a DCCH is located and there is a pointer to a PCCH, the MS tunes to the PCCH. After locking onto a PCCH, a GPRS-136 MS reads broadcast information and verifies the suitability of the PCCH in terms of signal strength and services. Once the MS has determined that the PCCH is suitable for service, it functions similar to a GSM GPRS MS. All of the functional messages sent to the network for packet operation are exactly the same as specified in GSM GPRS. The only difference is that the MS only attaches to the packet network using network mode I and does not perform combined attaches and location updates as specified in GSM [19]. On a request for an attach, authentication of the MS may be performed using GSM-defined procedures (i.e., the SGSN obtains triplet information and challenges the MS). If the MS passes authentication and encryption is supported, GSM encryption is used and subscriber data from the GPRS HLR is downloaded into the SGSN. After successful GSM procedures are executed, the MS registers for service on the TIA/EIA-136 network through the tunneling of TIA/EIA-136 layer 3 messages between the MS and the gateway/serving MSC via the SGSN. This service registration on the TIA/EIA-136 network also includes TIA/EIA-41-based authentication procedures. On registration with the TIA/EIA-136 network, subscriber data is downloaded from the TIA/EIA-41 HLR to the gateway/serving MSC. The gateway/serving MSC/visitor location register (VLR) keeps track whether the MS is available on the packet network or circuit network.

In order to access a data service, the user is first required to establish a PDP context with the network. This identifies to the network the type of data network to which the MS wishes to connect (e.g., X.25 or IP) and, in the case of IP, if a static or dynamic IP address is to be used. The IP address space may belong to either a TIA/EIA-136 service provider or another data network. In addition, the context identifies the point of interconnect to the data network (the GGSN) by using a unique APN. In the case of dynamic IP allocation, the GGSN or network behind the GGSN allocates the IP address.

A class B136 MS is capable of both originating and terminating TIA/EIA-136 circuit calls while attached to the packet network. When a circuit call is delivered to the MS's home MSC, the MSC queries the HLR through TIA/EIA-41 procedures to locate the MS, and the HLR in turn sends an TIA/EIA-41 routing request to the gateway/serving MSC's VLR. The gateway/serving MSC sends a GSM hard page to the SGSN, which in turn pages the MS through the BS. If the MS user chooses to accept the call, packet operation is suspended and the MS tunes to the DCCH where it sends an TIA/EIA-136 page response. Conventional TIA/EIA-136 call delivery procedures are then completed. At the end of the

call, the MS may once again tune to the PCCH and attach to the packet network. In a similar fashion, when an MS user decides to originate a call while attached to the packet network, the MS suspends operation on the packet network, tunes to a DCCH, and originates the call.

A class B136 MS attached to the packet network is capable of originating and terminating TIA/EIA-136 teleservices, such as SMS. When a teleservice message is delivered to the TIA/EIA-136 teleservice server (message center), the teleservice server queries the MS's HLR, which in turn queries the gateway/serving MSC's VLR to request delivery of the teleservice. When the teleservice server is notified that the MS is located in the gateway/serving MSC's service area, the teleservice message is sent to the gateway/serving MSC. The gateway/serving MSC repackages the teleservice message into the appropriate TIA/EIA-136 layer 3 message (R-DATA) and tunnels the message to the SGSN. On receipt of the tunneled message, the SGSN pages the MS and waits for a response from the MS before delivering the tunneled message across the air interface. The gateway/serving MSC may require the MS to authenticate prior to delivering the teleservice message for fraud control purposes. When an MS originates a teleservice, the appropriate TIA/EIA-136 teleservice message is tunneled from the MS to the gateway/serving MSC through the SGSN. The gateway/serving MSC then repackages the teleservice message into the appropriate TIA/EIA-41 message for delivery to the teleservice server.

The Evolution Path to High-Speed Data

To enable a smooth transition to mobile networks that are able to accommodate Internet/intranet and other multimedia services, the ITU has developed requirements for 3G wireless systems defined as International Mobile Telecommunications in the year 2000 (IMT-2000). UWC-136 is the UWCC 3G radio access technology that meets or exceeds all requirements specified by the ITU for IMT-2000. UWC-136 defines a three-step evolutionary path for the TIA/EIA-136 digital air-interface: current TIA/EIA-136 to 136+ to 136HS.

The UWC-136 proposal also supports a basic physical level of compatibility with GSM EDGE-based TDMA technology. This compatibility includes the harmonization and agreed-on convergence of using an 8-PSK modulation scheme for both 136HS and ETSI EDGE. The compatibility of 136HS and ETSI EDGE is envisioned to:

- Ease the production of terminals that could operate with two radio interface standards, thus facilitating global terminal roaming
- Increase the commonality in network infrastructure design (allowing common hardware and software components), thus improving equipment choice, flexibility, and cost reduction through competition and benefits of scale
- Allow similar network planning engineering rules, commonality of operation and maintenance systems, and so on, thus reducing the operational costs of service providers who may operate both systems
- Encourage the continued growth of a global industry with efficient use of resources

These goals are realizable with the UWCC and ETSI approval of a cooperative agreement to strengthen and broaden the role of TDMA technologies for 3G systems.

UWC-136 enables user data rates varying from 64 to greater than 384 kb/s in all defined environments. In indoor office environments, it provides user data rates that exceed 2 Mb/s. In each environment, advanced link adaptation algorithms adjust modulation while variable channel coding ensures optimal user data rates for the available signal quality.

The 136HS air interface was developed to exceed the requirements for an IMT-2000 RTT while also incorporating requirements for a commercially effective evolution and deployment in current TIA/EIA-136 networks. Such considerations include flexible spectrum allocation, spectrum efficiency, compatibility with current TIA/EIA-136 and 136+, and support of macrocellular performance at higher mobile speeds. 136HS complements the current TIA/EIA-136 and future 136+ systems, and is the high-speed component of UWC-136. 136HS is designed to exceed the IMT-2000 data services requirements for all defined environments: high-speed vehicular, low-speed vehicular, pedestrian/outdoor, and indoor office.

As previously stated, 136HS systems use a 200 kHz RF carrier with physical-layer compatibility with GSM EDGE-based technology to provide user data rates that exceed 384 kb/s. Advanced resource and network management schemes allow 136HS 200 kHz channels to be deployed in a 1/3 reuse pattern. This reuse capability will ensure that an initial 136HS system can be deployed in less than 1 MHz of spectrum. Thus, 136HS minimizes the impact on service providers evolving to IMT-2000 service while preserving the spectrum investment. For indoor office environments, 136HS uses a 1.6 MHz RF channel to provide user data rates which exceed 2 Mb/s.

UWC-136 supports a service protocol that is band-transparent across multispectral bands (i.e., 800 MHz and 1900 MHz) which are commercially operating today. It is important to note that the technologies submitted as UWC-136 for IMT-2000 are band-independent and may be deployed by existing or new operators. UWC-136 also provides a logical extension and enhancement of current TIA/EIA-136 features into new spectral bands as they become available. Such incremental new features as high-speed data and multimedia services can be introduced in a manner compatible with the ongoing demand for existing TIA/EIA-136 services. In addition, these features can be focused on the precise market segments requiring such additional capabilities.

The 3G specifications contained in the UWC-136 RTT is compatible with second-generation technology at the core network level, which will ensure a seamless transition to 3G technology. Modernizing networks to handle asynchronous transfer mode and packet data ensures that the core networks will be 3G-capable and that only the air interface will need enhancement to support 3G service requirements.

Conclusions

In the last year, the IS-136 community put into place a three-part strategy for evolving its technology toward third-generation IMT-2000. The strategy consists of enhancing the voice and data capabilities of the 30 kHz channels, while adding the evolved GSM 200 kHz carrier for 384 kb/s data (EDGE), and adopting a wider 1.6 MHz bandwidth TDMA carrier for 2 Mb/s data. This article has described the physical layer improvements to the 30 kHz carrier in terms of the new 8-PSK modulation, new slot formats, and the applications that come with the next revision of IS-136 to be called ANSI 136 Revision A. ACELP/CC2 with three-slot interleaving for improved robustness was shown to give an additional 3.7 dB performance improvement, while a new vocoder equivalent to GSM enhanced full-rate was also added which is ideal for wireless office type applications. In addition, new reporting mechanisms for BER, FER, and real-time C/I were described which will significantly enhance the capability of the system to characterize its environment. Finally, several new projects that are underway within UWCC or TIA were described, such as the ability to support six users per RF carrier (TDMA6), downlink time diversity, and fast power control. Although IS-136 has not been as visible as IS-95 or GSM in the media, it continues to be the digital wireless standard with the largest number of subscribers in the Americas. It is believed

that with these radio link advancements, in conjunction with the service enhancements also being developed, IS-136 is well poised for competition far into the next century.

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Biographies

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CAMERON COURSEY [M] is a senior member of technical staff at SBC Technology Resources, Inc., where he is responsible for the development of digital cellular features and services. He received his B.S. and M.S. degrees in electrical engineering from the University of Missouri-Rolla in 1987 and 1988, respectively. At SBC his primary activities have been the development of technical specifications for digital mobiles, acceptance testing of digital mobiles, development of software programs for digital network testing and optimization, development and implementation of test plans for the acceptance testing of digital mobiles and networks, development and standardization of intelligent roaming algorithms and non-public services, representation of SBC in the Global TDMA Forum (GTF) of the UWCC, chairmanship of the 136+ Working Group of the GTF directing the development of next-generation digital voice and data services, and representation of SBC in the Telecommunications Industry Association (TIA) TR-45.3 engineering subcommittee, the digital standards development organization. He is the author of the textbook *Understanding Digital PCS: The TDMA Standard*, Artech House, 1999.

PAUL HARTMAN is a senior member of technical staff at SBC Technology Resources, Inc., where he is responsible for the development of digital cellular features and services. He received his B.S. degree in electrical engineering from Southern Methodist University in 1983. Since joining SBC in 1997, his primary activities have been the development of technical specifications and requirements for vendor product development, development and implementation of test plans for digital network testing and optimization, and representation of SBC in the GTF of the UWCC. Prior to joining SBC, he was employed at Rockwell International from 1983 to 1987 in a technical position for the development of digital microwave radios and at Ericsson, Inc., from 1987 to 1997 in technical and product management positions related to the development and deployment of digital wireless technologies.

RICHARD KOBYLINSKI [M] is a senior member of technical staff at SBC Technology Resources, Inc. He received his B.S. and M.S. degrees in electrical engineering from the University of Missouri-Rolla in 1975 and 1977, respectively. He has close to 20 years of professional experience in the analysis of wireless communication systems. For 12 of those years he was employed by the McDonnell Douglas Corporation, where he was involved in the design and development of optical communication systems for satellite applications, and radio communication links for real-time missile control. For the last seven years he has been with SBC Technology Resources, Inc., which is the applied research and development subsidiary for the SBC family of companies. His major interests include both link- and system-level simulation as applied to cellular and PCS systems. He has most recently been involved in the UWCC GTF and TIA TR-45.3 engineering subcommittee in the development of the 136+ standard.

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KRISTER RAITH received an M.S.E.E. degree from Royal Institute of Technology, Stockholm, Sweden, in 1982. He started working at the Radio Research Department of Ericsson Radio Systems in Sweden in 1983 where he contributed to the early research in signal processing for digital cellular at Ericsson. He participated in the design and development of pre-GSM TDMA testbeds Ericsson Radio developed which culminated in the GSM Paris trials in 1986 and the subsequent adoption of TDMA. In 1988 he assumed responsibility for the digital signal processing for the Ericsson testbed for U.S. digital cellular which was successfully demonstrated in 1988. Since 1989 he has actively worked in TIA standards development and was a key contributor to the IS-54 standard and later to the evolution toward ANSI 136. He was involved in the UWCC deliberation of evolving ANSI 136 to the third generation, and was a key proponent of using EDGE for the physical layer which was subsequently adopted by UWC. Currently he is with Ericsson in Research Triangle Park, North Carolina, as manager of the Ericsson TIA air-interface group.

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